

Matrix inversion using multiple-input multiple-output adaptive filtering

Muhammad Yasir Siddique Anjum¹, Javed Iqbal²

¹Department of Electrical Engineering, Faculty of Engineering and Computing, National University of Modern Languages, Islamabad, Pakistan

²Department of Electrical Engineering, Military College of Signals, National University of Science & Technology, Islamabad, Pakistan

Article Info

Article history:

Received Apr 20, 2024

Revised Jan 10, 2025

Accepted Feb 20, 2025

Keywords:

Adaptive

Inversion

Matrix

Multiple-input multiple-output

Non-stationary

Normalized least mean squares

ABSTRACT

A new approach for matrix inversion is introduced. The approach is based on vector representation of multiple-input multiple-output (MIMO) channel matrix, in which the channel matrix is described by a linear combination of channel vectors weighted by their respective system inputs. The MIMO system output is then fed into a bank of adaptive filters, wherein the response of a given adaptive filter is iteratively minimized to match its output to the given system input. In doing so, adaptive filters equalize the impact of respective channel vectors on the MIMO channel output, while simultaneously orthogonalizing themselves from all other channel vectors, forming the channel matrix inverse. The method demonstrates satisfactory convergence and accuracy in Monte Carlo simulations conducted with varying signal-to-noise ratios (SNRs) and matrix conditioning scenarios. The suggested approach, by virtue of its adaptable characteristics, can also be employed for time-varying linear equation systems.

This is an open access article under the [CC BY-SA](https://creativecommons.org/licenses/by-sa/4.0/) license.



Corresponding Author:

Muhammad Yasir Siddique Anjum

Department of Electrical Engineering, Faculty of Engineering and Computing

National University of Modern Languages

Sector H-9, Islamabad Capital Territory 44000, Pakistan

Email: yasir.siddique@numl.edu.pk

1. INTRODUCTION

Adaptive filtering is an important branch of digital signal processing. Over the years, it has found applications in many areas of science and engineering, including mobile telecommunication systems [1]–[3], deep learning systems [4], [5], radar systems [6], [7], geophysics [8], [9], radiology [10], [11], image processing [12], speech applications [13]–[16], and power engineering [17]–[19]. This can be attributed to the advantages of adaptive filters in terms of real-time adaptation, noise rejection, robustness, convergence speed, and efficiency [20]. However, the application of adaptive filtering has been largely restricted to single-input single-output systems.

Multiple-input multiple-output (MIMO) systems arise in important areas of science and engineering, e.g., MIMO wireless communication systems [21], MIMO control systems [22], and MIMO acoustic systems [23]. MIMO systems can be represented by, or reduced to, a set of linear equations that describe the system response to multiple inputs. The system of linear equations needs to be solved in order to perform the necessary function. Adaptive filters can leverage their versatility to MIMO systems in solving such system of linear equations by computing the inverse of the MIMO channel matrix. This work provides a proof-of-concept by presenting a novel iterative method, based on a bank of adaptive filters, for matrix inversion in the context of MIMO wireless communication systems.

The method is based on the proposed vector representation of the MIMO channel matrix, wherein the channel matrix is described by a linear combination of channel vectors weighted by their respective

system inputs. The system output can then be fed into a bank of adaptive filters, where the response of a given adaptive filter is iteratively minimized to match its output to the given system input. In doing so, a given adaptive filter equalizes the impact of the respective channel vector on the MIMO channel output, while simultaneously orthogonalizing itself from all other channel vectors. The collective response vectors of all adaptive filters form the channel matrix inverse.

The proposed method is novel. As a proof of concept, its efficacy is evaluated via Monte Carlo simulations for a range of signal-to-noise ratio (SNR) and matrix conditions. Simulation results show that the proposed method achieves good convergence and accuracy properties. Further, the method has an added advantage that, due to its adaptive ability, it can update the matrix inverse if the system drifts over time, for example, in non-stationary channel conditions [24].

2. THEORY

2.1. System model

Consider a MIMO system with dimensions $N_T \times N_R$, with N_T representing the number of transmit nodes and N_R the number of receive nodes, such that $N_T = N_R$. The input signal vector $\mathbf{x}[n]$ passes through the channel and is multiplied by a channel matrix $\mathbf{H}$ to generate the output signal vector $\mathbf{y}[n]$.

$$\mathbf{y}[n] = \mathbf{H}\mathbf{x}[n] + \mathbf{v}[n] \quad (1)$$

Where $\mathbf{x}[n] = [x_0(n) \ x_1(n) \ \dots \ x_{N-1}(n)]^T$, $\mathbf{y}[n] = [y_0(n) \ y_1(n) \ \dots \ y_{N-1}(n)]^T$, and $\mathbf{v}[n] = [v_0(n) \ v_1(n) \ \dots \ v_{N-1}(n)]^T$ are column vectors of dimension $N \times 1$, with $x_i(n)$ and $y_i(n)$ respectively indicating the signal transmitted on and received at the i -th node at the time instant n . $\mathbf{v}[n]$ represents the noise vector with statistical characterization $N(0, \sigma^2 \mathbf{I})$ with $\mathbf{I}$ being the identity matrix. The channel matrix $\mathbf{H}$ is of dimensions of $N \times N$, and is described as (2):

$$\mathbf{H} = \begin{bmatrix} h_{00} & h_{01} & \dots & h_{0(N-1)} \\ h_{10} & h_{11} & \dots & h_{1(N-1)} \\ \vdots & \vdots & \ddots & \vdots \\ h_{(N-1)0} & h_{(N-1)1} & \dots & h_{(N-1)(N-1)} \end{bmatrix} \quad (2)$$

In (1) can be rearranged as (3):

$$\mathbf{y}[n] = \sum_{i=0}^{N-1} \begin{bmatrix} h_{0i} \\ h_{1i} \\ \vdots \\ h_{(N-1)i} \end{bmatrix} x_i(n) + \mathbf{v}[n] \quad (3)$$

with $i = 0, 1, \dots, N-1$. An estimate of $x_i(n)$ can be iteratively obtained from $\mathbf{y}[n]$ by using an adaptive filter (Figure 1). N such adaptive filter blocks can be used to obtain an estimate of $\mathbf{x}[n]$ from $\mathbf{y}[n]$, with filter weights of the blocks minimizing the impact of channel matrix $\mathbf{H}$ on $\mathbf{y}[n]$ by effectively acting as an inverse of $\mathbf{H}$.

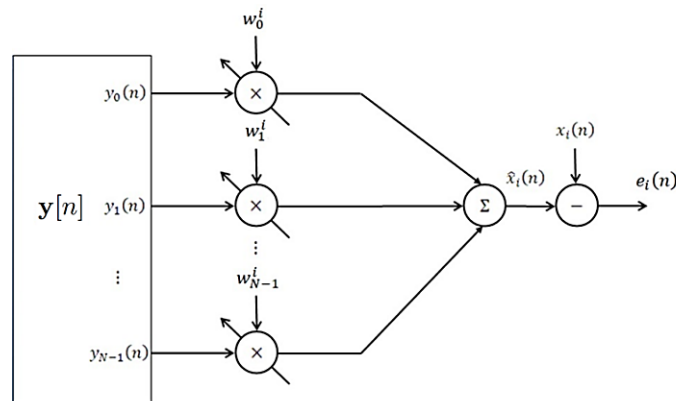


Figure 1. The i -th adaptive filter block for computation of matrix inverse. There are N such blocks. N represents the square matrix size

2.2. MIMO adaptive filter

For given value of i , in order to obtain an estimate $\hat{x}_i(n)$ of $x_i(n)$, an adaptive filter block (Figure 1) can be created with a weight vector $w^i = [w_0^i, w_1^i, \dots, w_{N-1}^i]^T$, so that the i -th adaptive filter output $x_i(n)$ is:

$$\hat{x}_i(n) = w^{iT} y[n] \quad (4)$$

Mean squared error (MSE) between the filter output $\hat{x}_i(n)$ and the desired output $x_i(n)$ is defined as (5) [25]:

$$E\{e_i^2(n)\} = \sigma_{x_i}^2 - 2w^{iT} p_i + w^{iT} R w^i \quad (5)$$

with error $e_i(n) = x_i(n) - \hat{x}_i(n)$. $\sigma_{x_i}^2 = E\{x_i^2(n)\}$ represents the power of the i -th desired signal $x_i(n)$. $N \times 1$ cross-correlation vector, p_i , between filter input and desire output is defined as (6):

$$p_i = [p_{0i}(n) \quad p_{1i}(n) \quad \dots \quad p_{(N-1)i}(n)] \quad (6)$$

with $p_{ji}(n) = E\{y_j(n)x_i(n)\}$ and $j = 0, 1, \dots, N-1$. $N \times N$ autocorrelation matrix R of the filter input $y[n]$ is defined as (7):

$$R = \begin{bmatrix} r_{00}(n) & r_{01}(n) & \dots & r_{0(N-1)}(n) \\ r_{10}(n) & r_{11}(n) & \dots & r_{1(N-1)}(n) \\ \vdots & \vdots & \ddots & \vdots \\ r_{(N-1)0}(n) & r_{(N-1)1}(n) & \dots & r_{(N-1)(N-1)}(n) \end{bmatrix} \quad (7)$$

such that $r_{ij}(n) = E\{y_i(n)y_j(n)\}$. Gradient of MSE taken with respect to the filter weights, ∇ , leads to the Wiener-Hopf equation [25]:

$$w_{\text{opt}}^i = R^{-1} p_i \quad (8)$$

with ∇ being:

$$\nabla = -2p_i + R w^i \quad (9)$$

For $i = 0, 1, \dots, N-1$:

$$W_{\text{opt}}^T = R^{-1} P \quad (10)$$

such that $W_{\text{opt}}^T = [w_{\text{opt}}^0, w_{\text{opt}}^1, \dots, w_{\text{opt}}^{N-1}]$ and $P_{\text{opt}} = [p_0, p_1, \dots, p_{N-1}]$. W , which attempts to achieve a minimum MSE estimate of $x[n]$ from $y[n]$ by minimizing the impact of H on the latter, effectively acts as an inverse of H . Iterative computation of W_{opt} is discussed in next section.

2.3. Iterative implementation

A starting choice for computing W_{opt} iteratively can be the Steepest Descent algorithm [25] due to the simplicity of the form:

$$w^i[n+1] = w^i[n] - \mu \nabla \quad (11)$$

μ represents algorithm's iteration step-size. Substituting the value of the gradient from (9) in (11), and rearranging leads to:

$$w^i[n+1] = w^i[n] + 2\mu y[n] e_i(n) \quad (12)$$

In (12) represents famous least mean squares (LMS) algorithm [25]. A concern in the implementation of LMS algorithm is the choice of μ . Large value of μ may lead to faster convergence but poor accuracy. Small value of μ may lead to higher accuracy but slower convergence. One way to overcome the issue is to make μ inversely proportional to the input signal energy, which in this case is $y^T[n]y[n]$. This leads to the normalized least mean squares (NLMS) algorithm [25]:

$$w^i[n+1] = w^i[n] + \frac{1}{y^T[n]y[n]} y[n] e_i(n) \quad (13)$$

For $i = 0, 1, \dots, N-1$,

$$W[n+1] = W[n] + \frac{1}{y^T[n]y[n]} y[n]e^T[n] \quad (14)$$

with $e[n] = [e_0(n), e_1(n), \dots, e_{N-1}(n)]^T$.

3. METHODS

Simulations were performed for determining the efficacy of the proposed method. A 5×5 autocorrelation matrix of the first-order autoregressive process, AR(1), is used for simulating the MIMO channel matrix H . A 5×5 matrix results in five adaptive filter blocks, i.e., $i = 0, 1, \dots, 4$. Five type of simulations were performed. In the first stage, filter weights $w^i[n]$ in (13), representing first-row estimates of W , were computed, overlaid and displayed as a function of the iterations performed by the NLMS algorithm as shown in Figure 2. In the second stage, square of the error term $e[n]$ in (14), representing the individual error estimates between the respective desired outputs $x_i(n)$ and the estimated outputs $\hat{x}_i(n)$, were computed, overlaid and presented as function of NLMS iterations as presented in Figure 3. In the third stage, element-wise MSE between respective rows of $W[n]$ in (14) and those of an exact matrix inverse were computed, overlaid and displayed as a function of number of NLMS iterations displayed in Figure 4. First, second, and third stage simulations were performed at SNR of 60 dB and one Monte Carlo noise run. In the fourth stage, element-wise MSE between the $W[n]$ in (14) and the exact matrix inverse was computed and displayed as function of SNR ranging from 0 to 100 dB in the increments of unity, with $n = 110$ and 100 Monte Carlo runs for noise simulation as shown in Figure 5. First four simulations were carried out with $\alpha = 0.4$. In the fifth stage, four simulations were performed for α ranging from 0.2 to 0.8 in increments of 0.2; results were overlaid and displayed in Figure 6. All simulations were performed in MATLAB (Mathworks, Natick, MA, USA) using in-house written scripts.

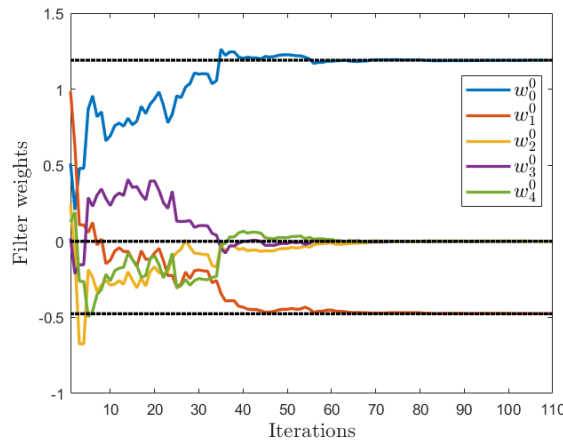


Figure 2. Filter weights $w^i[n]$ in (13) with $i = 0$, depicting first-row estimates of inverse matrix W

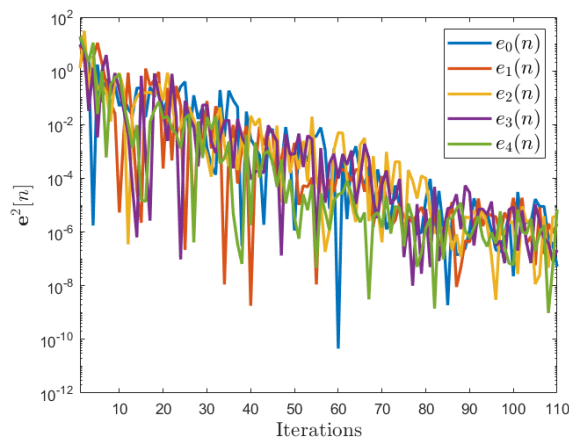


Figure 3. Squared NLMS-error term $e[n]$ in (14) showing algorithm convergence properties

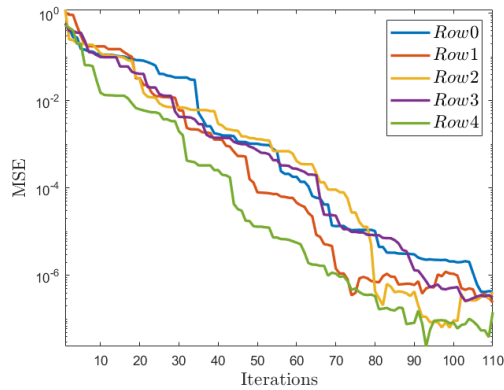


Figure 4. Element-wise MSE between respective rows of W and H^{-1}

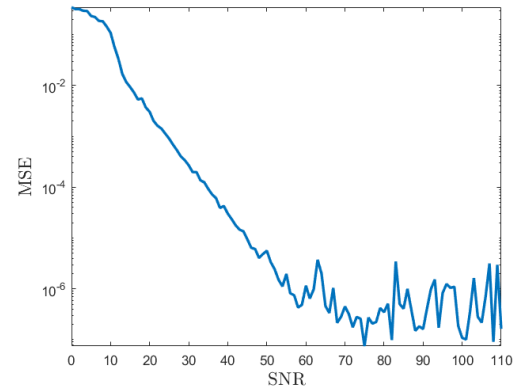


Figure 5. Element-wise MSE between W and H^{-1} , computed for a range of SNR values

4. RESULTS AND DISCUSSION

Figure 2 displays rapid convergence properties of representative estimates of the first adaptive filter block. Adaptive filter estimates converge to actual values in nearly less than seventy iterations. Also, once the estimates converge, they do not diverge or oscillate around the converged values. Figure 3 shows the row-wise squared NLMS-error, which demonstrates that the NLMS, employed as an optimization algorithm to compute the estimates, remains stable and does not diverge or oscillate. This can be attributed to the ability of the NLMS algorithm to adjust its step-size according to the input signal energy [25]. NLMS reduces the step-size to avoid the gradient noise, if the error is small; and if the error is large, NLMS increases the step-size to avoid convergence lag [25]. In addition, the instantaneous squared error reaches a value of 10^{-6} at the end of the eightieth iteration. This can be visualized in Figure 4, for all rows of the computed inverse matrix, wherein the element-wise MSE of the rows reaches the threshold of 10^{-6} , also at around the eightieth iteration. Figures 2 to 4 show the convergence and accuracy properties of the algorithm at SNR=60 dB, whereas Figure 5 shows the convergence and accuracy properties of the method for an SNR range of 0-110 dB. Up to 60 dB, the algorithm displays a negative log-linear trend between the MSE and the SNR, and the MSE enters a steady-state at nearly around 60 dB, without displaying any further reduction. This behavior becomes marked in Figure 6, where the results are overlaid for different condition numbers of H . For $\alpha=0.2$, where the condition number of H is 2.01, MSE continues to decrease log-linearly with increase in SNR, reaching nearly 10^{-10} at 110 dB. On the other hand, for $\alpha=0.6$ (condition number 9.74), MSE not only reaches a steady state comparatively early, that is, at 30 dB, but also achieves a higher steady state value of 10^{-2} ; this trend continues for $\alpha=0.8$, where the condition number increases to 29.76. At $\alpha=1$, H becomes singular.

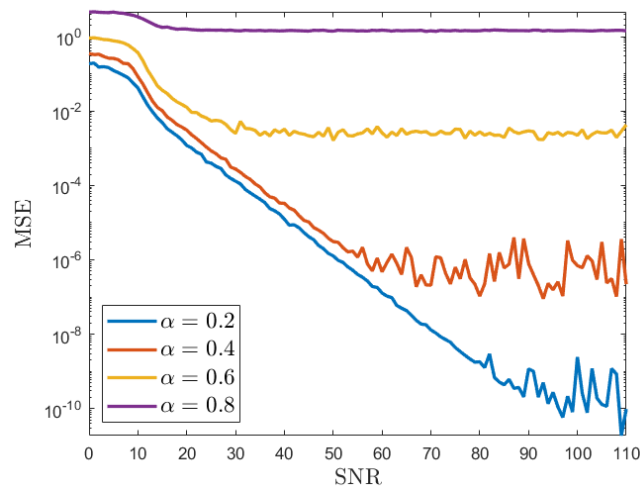


Figure 6. Element-wise MSE between W and H^{-1} , computed for range of matrix conditions

5. CONCLUSION

A novel matrix inversion method based on MIMO adaptive filtering is presented. Monte Carlo simulation results demonstrate that the method has good convergence and accuracy properties. The proposed method has the ability to iteratively adapt to changes in system conditions, which makes it suitable for practical implementation in systems with time-varying properties, e.g., MIMO wireless systems, MIMO acoustic systems, and MIMO control systems.

FUNDING INFORMATION

No funding is involved.

AUTHOR CONTRIBUTIONS STATEMENT

This journal uses the Contributor Roles Taxonomy (CRediT) to recognize individual author contributions, reduce authorship disputes, and facilitate collaboration.

Name of Author	C	M	So	Va	Fo	I	R	D	O	E	Vi	Su	P	Fu
Muhammad Yasir Siddique Anjum	✓	✓	✓	✓	✓	✓		✓	✓	✓			✓	
Javed Iqbal	✓	✓				✓		✓		✓		✓		

C : Conceptualization

M : Methodology

So : Software

Va : Validation

Fo : Formal analysis

I : Investigation

R : Resources

D : Data Curation

O : Writing - Original Draft

E : Writing - Review & Editing

Vi : Visualization

Su : Supervision

P : Project administration

Fu : Funding acquisition

CONFLICT OF INTEREST STATEMENT

Authors state no conflict of interest.

DATA AVAILABILITY

Data availability is not applicable to this paper as no new data were created or analyzed in this study.

REFERENCES

- [1] J. E. Paul, "Adaptive digital techniques for audio noise cancellation," *IEEE Circuits & Systems Magazine*, vol. 1, no. 4, pp. 2–7, 1979, doi: 10.1109/MCAS.1979.6323383.
- [2] T. Bahraini and A. N. Sadigh, "Proposing a robust rls based subband adaptive filtering for audio noise cancellation," *Applied Acoustics*, vol. 216, 2024, doi: 10.1016/j.apacoust.2023.109755.
- [3] V. B. Shukla, O. Krejcar, K. Choi, A. K. Mishra, and V. Bhatia, "Adaptive sparse channel estimator for irs-assisted mmwave hybrid mimo system," *IEEE Transactions on Cognitive Communications and Networking*, vol. 10, no. 6, pp. 2224–2235, 2024, doi: 10.1109/TCCN.2024.3422510.
- [4] M. H. Al-Meer, "MRI acoustic noise reduction using cnn," *Journal of Engineering Research*, pp. 1–17, 2022, doi: 10.36909/jer.17661.
- [5] Q. Zhang, J. Li, Y. Sun, S. Wang, J. Gao, and B. Yin, "Beyond low-pass filtering on large-scale graphs via adaptive filtering graph neural networks," *Neural Networks*, vol. 169, pp. 1–10, 2024, doi: 10.1016/j.neunet.2023.09.042.
- [6] A. Khwairakpam, D. Kandar, and B. Paul, "Noise reduction in synthetic aperture radar images using fuzzy logic and genetic algorithm," *Microsystem Technologies*, vol. 25, no. 5, pp. 1743–1752, 2019, doi: 10.1007/s00542-017-3474-x.
- [7] F. Jin and S. Cao, "Automotive radar interference mitigation using adaptive noise canceller," *IEEE Transactions on Vehicular Technology*, vol. 68, no. 4, pp. 3747–3754, 2019, doi: 10.1109/TVT.2019.2901493.
- [8] M. Hattingh, "A new data adaptive filtering program to remove noise from geophysical time- or space-series data," *Computers and Geosciences*, vol. 14, no. 4, pp. 467–480, 1988, doi: 10.1016/0098-3004(88)90029-5.
- [9] Z. Wang, G. Liu, C. Li, L. Shi, and Z. Wang, "Random noise attenuation of 3d multicomponent seismic data using a fast adaptive prediction filter," *Geophysics*, vol. 89, no. 3, pp. V263–V280, 2024, doi: 10.1190/geo2023-0195.1.
- [10] A. Mincholé, J. Camps, A. Lyon, and B. Rodríguez, "Machine learning in the electrocardiogram," *Journal of Electrocardiology*, vol. 57, pp. S61–S64, 2019.
- [11] N. V. Thakor and Y. S. Zhu, "Applications of adaptive filtering to eeg analysis: noise cancellation and arrhythmia detection," *IEEE Transactions on Biomedical Engineering*, vol. 38, no. 8, pp. 785–794, 1991, doi: 10.1109/10.83591.
- [12] M. Su, J. Zheng, Y. Yang, and Q. Wu, "A new multipath mitigation method based on adaptive thresholding wavelet denoising and double reference shift strategy," *GPS Solutions*, vol. 22, no. 2, pp. 1–12, 2018, doi: 10.1007/s10291-018-0708-z.
- [13] A. Pandey and D. Wang, "A new framework for cnn-based speech enhancement in the time domain," *IEEE/ACM Transactions on Audio Speech and Language Processing*, vol. 27, no. 7, pp. 1179–1188, 2019, doi: 10.1109/TASLP.2019.2913512.
- [14] K. Audhkhasi, O. Osoba, and B. Kosko, "Noise-enhanced convolutional neural networks," *Neural Networks*, vol. 78, pp. 15–23, 2016, doi: 10.1016/j.neunet.2015.09.014.

- [15] N. Howard, A. Park, T. Z. Shabestary, A. Gruenstein, and R. Prabhavalkar, "A neural acoustic echo canceller optimized using an automatic speech recognizer and large scale synthetic data," in *ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing - Proceedings*, 2021, pp. 7128–7132, doi: 10.1109/ICASSP39728.2021.9413585.
- [16] D. Kim, H. Choi, and H. Bae, "Acoustic echo cancellation using blind source separation," in *IEEE Workshop on Signal Processing Systems, SIPS: Design and Implementation*, 2003, pp. 241–244, doi: 10.1109/SIPS.2003.1235676.
- [17] K. Belalia, M. Khodja, H. Bouzeboudja, A. Bendiabdellah, and A. Mostefa, "Network current quality enhancement under nonlinear and unbalanced load conditions using a four-wire inverter-based active shunt filter," *Indonesian Journal of Electrical Engineering and Informatics*, vol. 9, no. 3, pp. 601–614, 2021, doi: 10.52549/V9I3.2951.
- [18] P. K. Meher and S. Y. Park, "Area-delay-power efficient fixed-point lms adaptive filter with low adaptation-delay," *IEEE Transactions on Very Large Scale Integration (VLSI) Systems*, vol. 22, no. 2, pp. 362–371, 2014, doi: 10.1109/TVLSI.2013.2239321.
- [19] W. Al-Saedi, S. W. Lachowicz, D. Habibi, and O. Bass, "Power quality enhancement in autonomous microgrid operation using particle swarm optimization," *International Journal of Electrical Power and Energy Systems*, vol. 42, no. 1, pp. 139–149, 2012, doi: 10.1016/j.ijepes.2012.04.007.
- [20] J. M. McCool and B. Widrow, "Principles and applications of adaptive filters-a tutorial review," *IEEE International Symposium on Circuits and Systems*, vol. 3, pp. 1143–1157, 1980.
- [21] M. Khurana, C. Ramakrishna, and S. N. Panda, "Capacity enhancement using mu-mimo in vehicular ad hoc network," *International Journal of Applied Engineering Research*, vol. 12, no. 16, pp. 5872–5883, 2017.
- [22] M. Mudassar and A. Mujahid, "Modeling and fabrication of smart robotic wheelchair instructed by head gesture," *Indonesian Journal of Electrical Engineering and Informatics*, vol. 9, no. 3, pp. 633–646, 2021, doi: 10.52549/V9I3.2892.
- [23] Y. Huang, J. Benesty, and J. Chen, "Identification of acoustic mimo systems: challenges and opportunities," *Signal Processing*, vol. 86, no. 6, pp. 1278–1295, 2006, doi: 10.1016/j.sigpro.2005.06.023.
- [24] T. S. Rappaport, *Wireless communications: principles and practice*. Upper Saddle River, United States: Prentice Hall, 2024.
- [25] B. Farhang-Boroujeny, *Adaptive filters: theory and applications*. Hoboken, USA: John Wiley & Sons, 2003.

BIOGRAPHY OF AUTHOR



Muhammad Yasir Siddique Anjum    received bachelor's degree in electrical engineering from the Federal Urdu University of Arts, Science & Technology, Islamabad, Pakistan, in 2008; and a master's degree also in electrical engineering from the University of Engineering and Technology, Taxila, Pakistan, in 2015. Currently, he is working as a Lecturer at the Department of Electrical Engineering, National University of Modern Languages, Islamabad, Pakistan. His research interests include adaptive and random signal processing for wireless communication systems. He can be contacted at email: yasir.siddique@numl.edu.pk.



Javed Iqbal    received bachelor's degree in electronic engineering from International Islamic University, Islamabad, Pakistan, in 2010; and master's and PhD degrees, respectively, in telecommunications engineering and electrical, electronics, and telecommunications engineering from Politecnico di Torino, Italy, in 2014 and 2018. Currently, he is working as Assistant Professor at the Department of Electrical Engineering, Military College of Signals, National university of Science & Technology, Islamabad, Pakistan. His research interests include digital design, applied ai, internet of things, signal processing. He can be contacted at email: javediqbal@mcs.edu.pk.